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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/019,450	05/28/2002	Ravi Chandran	2376.2115-014.	4359

7590 11/09/2007

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EXAMINER	
WOZNIAK, JAMES S	
ART UNIT	**PAPER NUMBER**
2626	
MAIL DATE	**DELIVERY MODE**
11/09/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/019,450	Applicant(s) CHANDRAN ET AL.	
	Examiner James S. Wozniak	Art Unit 2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *15 August 2007*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-60* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-60* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *28 May 2002* is/are: a)☐ accepted or b)☒ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. In response to the office action from 2/15/2007, the applicant has submitted a request for continued examination, filed 8/15/2007, amending independent claims 1, 25, 31, and 55, while arguing to traverse the art rejection based on the allegedly claimed feedback loop *(Amendment, Pages 20-22)* and the use of a hybrid coded-linear format *(Amendment, Page 23)*. Applicant's arguments have been fully considered, however the previous rejection is maintained due to the reasons listed below in the response to arguments.

2. In response to the previous 35 U.S.C. 101 rejection the applicants argue that claims 25 and 55 now feature a step for "transmitting", which overcomes the corresponding 35 U.S.C. 101 rejection *(Amendment, Page 19)*. The examiner notes that, in this case, the transmission of modified bits to a far end device to present the bits in audio form to a user would be a "useful, concrete, and tangible result", and thus, the aforementioned rejection has been withdrawn.

3. In response to the amended claims, the examiner has withdrawn the previous claim objections directed to minor informalities and the previous 35 U.S.C. 112, second paragraph rejection.

Response to Arguments

4. Applicant's arguments have been fully considered but they are not persuasive for the following reasons:

The applicants have amended claims 1, 25, and 55 to revert them to their previous format and argues that these claims would not be single means claims because they do not utilize the "means for" or "step for" language *(amendment, Pages 20).* First, it is noted that Claims 25 and 55 are not a single means/step claims because they have been further amended to include the additional step/module for "transmitting". In addressing the applicants' arguments regarding claim 1, the examiner points out that although the aforementioned claims do not expressly utilize "means for" language, they are nonetheless means-plus-function claims because the "processor" of claim 1 is described in terms of the function it performs *(Seal-Flex, Inc. v. Athletic Track and Court Construction, 172 F.3d 836, 850, 50 USPQ2d 1225, 1234 (Fed. Cir.1999) (Radar, J., concurring) ("claim elements without express step-plus-function language may nevertheless fall within 112 6 if they merely claim the underlying function without recitation of acts for performing that function)) (also see MPEP 2181).* Claim 1 recites a "processor" (means) to (for) "read...generate...derive...replace" (function), thus, claim 1 is in means-plus-function format and accordingly is a single means claim.

Now addressing the art rejections, the applicant's argue that Jarvinen et al *(U.S. Patent: 5,946,651)* fails to teach a feedback loop for feeding back an adjusted first parameter for parameter replacement and "the concept of adjusting a gain parameter over a network prior to reception at a receiver" *(Amendment, Page 21).*

In response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the features upon which applicant relies (i.e., a feedback loop or feeding back an adjusted parameter to further adjust the adjusted parameter) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993). It is pointed out that the current broad claim scope only specifies: "generate an adjusted first parameter value...and responsive to said adjusted first parameter to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter", and there is no mention of a feedback loop or "feeding back" an adjusted signal in the claims. As such, Jarvinen teaches that a gain factor corresponding to an excitation is adjusted with a scaling factor *(i.e., "generate an adjusted first parameter value", Col. 7, Lines 58- Col. 8, Line 61, Fig. 3- b is adjusted in Element 312)* and the generated scaled gain parameters are used to obtain *(i.e., derive, Fig. 3, derived adjusted gain signal p)* a scaled gain value in replacement of an original gain value to be applied to the corresponding excitation signal *(Col. 7, Line 58- Col. 8, Line 61)*. Also, it is worth pointing out that Jarvinen discloses further adjustment of a fed back gain scaled signal *(Col. 8, Line 62-67)*. Thus, for at least the preceding reasons, Jarvinen teaches the claimed "generate an adjusted first parameter value...and responsive to said adjusted first parameter to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter". Furthermore, in response to "the concept of adjusting a gain parameter over a network prior to reception at a receiver" argument, the examiner points out that such a concept is taught by Yajima et al *(U.S. Patent: 6,300,534)* in the form of speech parameter adjustment in the compressed domain at a relay node *(Col. 9, Line 35-*

Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53; and Figs. 13, and 17-18).

The applicant further provides a summary of the Yajima et al *(U.S. Patent: 6,300,534)* reference and argues that Yajima fails to teach any type of feedback technique as claimed by the applicants *(Amendment, Pages 21-22).* In response and as is noted above, it is pointed out that the presently claimed invention does not recite a feedback loop or feeding back of an already adjusted gain signal. Also, the limitation "generate an adjusted first parameter value...and responsive to said adjusted first parameter to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter" is taught by Jarvinen, as is pointed out above.

The applicant next argues that one of ordinary skill in the art would not seek to employ the improvements described by Yajima in Jarvinen because Jarvinen is synthesizing speech at a listener's device *(Amendment, Page 22).* In response, the examiner notes that combining Jarvinen and Yajima would allow for Jarvinen's perceptual enhancement in a system that could allow for connection between two different types of networks *(Yajima, Col. 3, Lines 25-27)*, efficient voice coding *(Yajima, Col. 7, Lines 25-32)*, and an inherent decrease in the amount of processing performed at a receiver. Furthermore, Jarvinen discloses modifying speech coding parameters before synthesis *(see Fig. 3)*, while Yajima teaches that a decoder can be used obtain speech parameters for modification through partial decoding *(Fig. 13, Element 108).* Thus, since one of ordinary skill in the art, looking to the teachings of Yajima, would recognize that parameters could be modified in the coded domain at a relay node instead of at a receiver (as in Jarvinen) for the aforementioned benefits, the combination of Jarvinen and Yajima is proper. In

response to the applicant's additional argument that Jarvinen "is not replacing adjusted parameters" *(Amendment, page 22)*, the examiner notes that the current claim language recites "replac[ing] said first parameter" not "said adjusted first parameter". In response to such an argument, please see the above response directed towards the feedback loop argument *(i.e., an original gain parameter b is replaced by a scaled gain parameter p)*.

Thus, for at least the aforementioned reasons, claims 1 and 31 remain rejected under U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima et al.

With respect to Claims 25 and 55, the applicants argue that Yajima fails to teach a network with a hybrid coded/linear format *(Amendment, Pages 22-23)*. In response, the examiner notes that such a coding format is well known in the prior art as is evidenced by the applicant's admitted prior art *("TFO standard", Pages 25-26)*. As such, the applicants' arguments have been fully considered, but are moot with respect to the new grounds of rejection in view of the applicant's admitted prior art.

The art rejections of the remainder of the dependent claims are traversed for reasons similar to the independent claims *(Amendment, Pages 22-24)*. In regards to such arguments, see the response directed to the appropriate independent claims.

Drawings

5. **Figures 1-2** should be designated by a legend such as --Prior Art-- because only that which is old is illustrated *(see Background of the Invention, Pages 1-4)*. See MPEP § 608.02(g). Corrected drawings in compliance with 37 CFR 1.121(d) are required in reply to the Office

action to avoid abandonment of the application. The replacement sheet(s) should be labeled "Replacement Sheet" in the page header (as per 37 CFR 1.84(c)) so as not to obstruct any portion of the drawing figures. If the changes are not accepted by the examiner, the applicant will be notified and informed of any required corrective action in the next Office action. The objection to the drawings will not be held in abeyance.

Claim Rejections - 35 USC § 112

6. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

7. **Claims 1-24** are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter that was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention.

Claim 1 recites "a processor responsive to..." but lack means for performing the operations that the processor performs.

A single means claim, i.e., where a means recitation does not appear in combination with another recited element of means, is subject to an undue breadth rejection under 35 U.S.C. 112, first paragraph. In re Hyatt, 708 F.2d 712, 714-715, 218 USPQ 195, 197 (Fed. Cir. 1983) (A single means claim which covered every conceivable means for achieving the stated purpose was held nonenabling for the scope of the claim because the specification disclosed at most only

those means known to the inventor.). When claims depend on a recited property, a fact situation comparable to Hyatt is possible, where the claim covers every conceivable structure (means) for achieving the stated property (result) while the specification discloses at most only those known to the inventor.

Dependent claims 2-24 do not remedy the lack of enablement issue noted above with respect to claim 25, and therefore, are also rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement.

Claim Rejections - 35 USC § 103

8. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

9. **Claims 1-6, 10-11, 15, 24, 31-36, 38, 40-41, 45, and 54** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al (U.S. Patent: 5,946,651) in view of Yajima et al *(U.S. Patent: 5,873,058).*

With respect to **Claims 1 and 31**, Jarvinen discloses:

In a communications system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic *(receiving transmitted*

coded speech parameters at a decoder including LPC coefficients and a gain parameter, Col. 6, Lines 16-58) said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic, an apparatus for adjusting the first characteristic comprising:

A processor responsive to said digital signals to read at least said first parameter and to generate at least a first parameter value derived from said first parameter *(decoding excitation parameters having associated gain factors, Col. 6, Lines 16-58)*;

Responsive to said digital signals and said first parameter value to generate an adjusted first parameter value representing an adjustment of said first characteristic *(adjusting a gain factor with a scaling factor, Col. 7, Line 58- Col. 8, Line 61)*; and

Responsive to said adjusted first parameter value to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter *(replacing an excitation parameter and associated gain with a perceptually adjusted excitation parameter, Col. 7, Line 34- Col. 8, Line 61).*

Jarvinen does not teach the concept of adjusting a gain parameter over a network prior to reception at a receiver, however Yajima discloses the concept of speech signal gain parameter adjustment at a relay node *(Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53; and Figs. 13, partial decoder, Element 108 and 17-18).*

Jarvinen and Yajima are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen with the concept of gain adjustment at a relay node as taught by Yajima in order to implement gain

adjustment at a means that is capable of connecting two different types of networks *(Col. 3, Lines 25-27)* and coding voice efficiently *(Col. 7, Lines 25-32)*, while also inherently decreasing the amount of processing performed at a receiver..

With respect to **Claims 2 and 32**, Jarvinen discloses:

The first characteristic comprises a level of the audio signal *(gain factor that is indicative of a desired speech signal level, Col. 5, Line 25- Col. 6, Line 32; and Col. 12, Lines 24-33).*

With respect to **Claims 3 and 33**, Yajima further discloses:

Yajima teaches avoiding synthesizing filter processing for a normal voice signal that would not require gain adjustment *(Col. 22, Lines 1-24; and adjusting a gain speech parameter, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53).*

With respect to **Claims 4 and 34**, Yajima teaches synthesizing filter processing, as applied to Claims 3 and 33.

With respect to **Claims 5 and 35**, Jarvinen discloses:

The compression code comprises a linear predictive code *(LP coefficients, Col. 5, Lines 25-57).*

With respect to **Claims 6 and 36**, Jarvinen discloses:

The compression code comprises regular pulse excitation long term prediction code *(LTP prediction coefficients, Col. 5, Lines 25-57).*

With respect to **Claims 10 and 40**, Yajima further discloses gain adjustment implementation at a relay node situated on a network that would inherently be capable of receiving near and far end speech from various transmission nodes connected to the network

(Fig. 16, Element 404; Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53).

With respect to **Claims 11 and 41**, Jarvinen discloses:

The processor test the adjusted first parameter value for an overflow and underflow condition before deriving the adjusted first parameter *(multiple threshold comparisons, Col. 7, Line 58- Col. 8, Line 61).*

With respect to **Claims 15 and 45**, Jarvinen discloses performing the decoding processing, as applied to Claim 1, on a plurality of parameters from a series of time frames *(Col. 6, Lines 16-58; and Col. 12, Lines 52-54).*

With respect to **Claims 24 and 54**, Jarvinen further discloses:

The processor performs at least the first decoding step to generate decoded signals related to the first characteristic of the audio signal *(recovering speech parameters using a speech decoder, Col. 6, Lines 16-26).*

With respect to **Claim 38**, Jarvinen discloses the use of the CELP coding standard *(Col. 5, Lines 25-35).*

10. **Claims 8-9, 12, 16, 18, 20-23, 39, 42, 46, 48, and 50-53** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima, and further in view of Yasunaga et al *(U.S. Patent: 6,330,534).*

With respect to **Claim 8**, Jarvinen in view of Yajima discloses the speech decoding apparatus utilizing perceptual gain scaling, as applied to Claim 1. Jarvinen in view of Yajima

does not explicitly teach the use of the algebraic CELP coding standard, however Yasunaga teaches the use of said standard *(Col. 3, Lines 42-51)*.

Jarvinen, Yajima, and Yasunaga are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen in view of Yajima with the ACELP standard taught by Yasunaga in order to provide a standard that reduces the complexities of computing coding distortions *(Yasunaga, Col. 3, Lines 42-51)*.

With respect to **Claim 9**, Jarvinen further teaches the gain scaling factor as applied to claim 1.

With respect to **Claims 12 and 42**, Jarvinen in view of Yajima discloses the speech decoding apparatus utilizing perceptual gain scaling, as applied to Claims 11 and 41. Jarvinen in view of Yajima does not teach that a decoder derives an adjusted speech parameter by quantizing an adjusted speech parameter, however Yasunaga discloses a process for adjusting a gain factor applied to a speech parameter by quantizing an adjusted target speech parameter *(Col. 30, Line 42- Col. 31, Line 9)*.

Jarvinen, Yajima, and Yasunaga are analogous art because they are from a similar field of endeavor in speech decoding utilizing adaptive gain control. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen in view of Yajima with the gain adjusting process taught by Yasunaga in order to provide a means for minimizing a quantization error between target and decoded speech parameters *(Yasunaga, Col. 30, Line 42 - Col. 31, Line 9)*.

Claims 16 and 46 contain subject matter similar to Claims 12, 15, 42, and 45 and thus, are rejected for the same reasons.

Claims 18 and 48 contains subject matter similar to Claim 12, and thus, is rejected for the same reasons.

With respect to **Claims 20 and 50**, Yasunaga further discloses scalar quantization performed using a predetermined quantization table *(Col. 12, Lines 10-21).*

With respect to **Claims 21 and 51**, Yasunaga further discloses subframe-based speech processing *(Col. 1, Line 33- Col. 2, Line 9).*

With respect to **Claims 22 and 52**, Yasunaga further discloses:

The processor replaces the first parameter with the adjusted first parameter for a first subframe before processing a subframe following the first subframe *(adjusting gains of processing frames within a speech frame on a frame-by-frame basis, Col. 28, Lines 40-50; and Col. 30, Line 42- Col. 31, Line 9).*

With respect to **Claims 23 and 53**, Yasunaga further discloses adjusting a gain of a current processing frame based on a gain of a previous processing frame *(Col. 30, Line 42- Col. 31, Line 9)*, and subframe-based speech processing, as applied to Claims 21 and 51.

Claim 39 contains subject matter similar to Claims 9 and 21, and thus, is rejected for the same reasons.

11. **Claims 7 and 37** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima et al in view of Yasunaga et al *(U.S. Patent: 6,330,534)*, and further in view of Crouse et al *(U.S. Patent: 4,899,384).*

With respect to **Claims 7 and 37**, Jarvinen et al in view of Yajima and further in view of Yasunaga teaches the speech decoding apparatus utilizing gain scaling, subframe based processing, and quantization processing, as applied to Claims 6, 21, 36, and 51. Jarvinen et al in view of Yajima and further in view of Yasunaga does not specifically suggest utilizing a maximum absolute value of a speech parameter to derive a speech scaling factor, however Crouse teaches the use of such a value *(Col. 5, Lines 5-16)*.

Jarvinen, Yajima, Yasunaga, and Crouse are analogous art because they are from a similar field of endeavor in speech coding systems. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen et al in view of Yajima and further in view of Yasunaga with the maximum absolute value parameter taught by Crouse in order to implement a speech coded method having reduced peak information that is consistent with a desired speech output quality *(Crouse, Col. 4, Lines 1-11)*.

12. **Claims 13-14, 17, 19, 43-44, 47, and 49** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jarvinen et al in view of Yajima et al in view of Yasunaga et al, and further in view of Swaminathan et al *(U.S. Patent: 5,751,903)*.

With respect to **Claims 13, 17, 19, 43, 47, and 49,** Jarvinen et al in view of Yajima and further in view of Yasunaga teaches the speech decoding apparatus utilizing perceptual gain scaling and quantization processing, as applied to Claims 12, 16, 18, 42, 46, and 48. Jarvinen et al in view of Yajima and further in view of Yasunaga does not teach the use of differential scalar quantization, however Swaminathan discloses the use of such a quantization during speech coding *(Col. 10, Lines 48-56)*.

Jarvinen, Yajima, Yasunaga, and Swaminathan are analogous art because they are from a similar field of endeavor in speech coding systems. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Jarvinen et al in view of Yajima and further in view of Yasunaga with the differential scalar quantization taught by Swaminathan in order to implement a means for quantizing speech parameters that requires a reduced number of bits *(Swaminathan, Col. 8, Lines 65-98).*

With respect to **Claims 14 and 44**, Yasunaga further discloses the use of a feedback loop having a speech parameter quantizer *(Col. 30, Line 42- Col. 31, Line 9; and Fig. 16)*, while Swaminathan discloses the use of differential scalar quantization as applied to Claims 13 and 33.

13. **Claims 25-30 and 55-60** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yajima et al *(U.S. Patent: 5,873,058)* in view of in view of the Applicants' Admitted Prior Art *(AAPA).*

With respect to **Claims 25 and 55**, Yajima discloses:

A transmitter transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic *(transmission node that outputs a coded voice signal, Col. 16, lines 52-60; wherein voice parameters comprise CELP coded speech and associated gain data, Col. 1, Line 33- Col. 2, Line 7; and Col. 9, Lines 52-57)* wherein said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic *(decodable speech*

parameters including a step for extracting voice parameters from a voice code signal, Col. 9, Line 35- Col. 10, Line 25; Col. 21, Lines 39-50); and

A processor responsive to said second bits to adjust said first bits and said second bits, whereby said first characteristic is adjusted *(adjusting a gain speech parameter at a relay device, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53),*

Wherein the processor adjusts the first characteristic without decoding said compression code *(partial decoding of speech parameters, Col. 33, Lines 42-65).*

A transmitter to transmit digital signal with adjusted first bits and second bits to a device to produce a corresponding audible signal with the first characteristic in the adjusted state *(Fig. 13).*

Yajima does not explicitly recite the combination of a compression code and a linear code to express a speech signal, however, such a coding scheme is well known in the prior art as is evidenced by the AAPA. The AAPA recites a TFO GSM standard using a combination of coded speech and PCM bits *("TFO standard", Page 25, Line 21- Page 26, Line 4).*

Yajima and the AAPA are analogous art because they are from a similar field of endeavor in speech compression. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yajima with the TFO GSM standard recited in the AAPA in order allow Yajima's gain controller to comply with well-known cellular network standards *(AAPA, Page 25, Lines 21-22).*

With respect to **Claims 26 and 56**, AAPA recites:

The linear code comprises PCM code *(PCM samples, Page 26, Line 4).*

With respect to **Claims 27 and 57**, Yajima discloses:

The first characteristic comprises audio level *(gain parameter which is indicative of an audio level, Col. 9, Line 35- Col. 10, Line 25; Col. 27, Line 40- Col. 29, Line 10; Col. 31, Lines 12-53).*

With respect to **Claims 28 and 58**, the AAPA recites the TFO GSM standard as applied to Claims 26 and 57.

With respect to **Claims 29 and 59**, the AAPA further recites first bits comprising the two LSBs and second bits comprising 6 MSBs *(Page 26, Lines 2-3).*

With respect to **Claims 30 and 60**, the AAPA further recites the use of PCM code for the 6 MSBs *(Page 26, Lines 2-3).*

Conclusion

14. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

Nahumi *(U.S. Patent: 5,898,675)*- discloses that it is well known in the art to adjust gain values of an audio signal in the compressed domain.

Bhaskaran *("Mediaprocessing in the Compressed Domain," 1996)*- discloses modification of audio data in the compressed domain, namely level adjustment *(see Section 3).*

15. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (571) 272-7632. The examiner can normally be reached on M-Th, 7:30-5:00, F, 7:30-4, Off Alternate Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached at (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

James S. Wozniak
10/24/2007